Gy

UNITED STATES PATENT AND TRADEMARK OFFICE

UNITED STATES DEPARTMENT OF COMMERCE
United States Patent and Trademark Office
Address: COMMISSIONER FOR PATENTS
P.O. Box 1450
Alexandria, Virginia 22313-1450
www.uspto.gov

| APPLICATION NO. | FILING DATE | FIRST NAMED INVENTOR | ATTORNEY DOCKET NO. | CONFIRMATION NO. |
|---|---|---|---|---|
| 09/985,976 | 11/07/2001 | Leonard E. Cornelisse | 10494-49 | 4630 |

7590 02/21/2007

Bhupinder S. Randhawa
Bereskin & Parr
Box 401
40 King Street West
Toronto, ON M5H 3Y2
CANADA

| EXAMINER | |
|---|---|
| FLANDERS, ANDREW C | |
| **ART UNIT** | **PAPER NUMBER** |
| 2615 | |

| SHORTENED STATUTORY PERIOD OF RESPONSE | MAIL DATE | DELIVERY MODE |
|---|---|---|
| 3 MONTHS | 02/21/2007 | PAPER |

**Please find below and/or attached an Office communication concerning this application or proceeding.**

If NO period for reply is specified above, the maximum statutory period will apply and will expire 6 MONTHS from the mailing date of this communication.

| Office Action Summary | Application No. | Applicant(s) |
|---|---|---|
| | 09/985,976 | CORNELISSE, LEONARD E. |
| | Examiner | Art Unit |
| | Andrew C. Flanders | 2615 |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☒ Responsive to communication(s) filed on *15 December 2006*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-40,42,43 and 45-57* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-40,42,43 and 45-57* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on *18 January 2002* is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

## DETAILED ACTION

### *Continued Examination Under 37 CFR 1.114*

A request for continued examination under 37 CFR 1.114, including the fee set forth in 37 CFR 1.17(e), was filed in this application after final rejection. Since this application is eligible for continued examination under 37 CFR 1.114, and the fee set forth in 37 CFR 1.17(e) has been timely paid, the finality of the previous Office action has been withdrawn pursuant to 37 CFR 1.114. Applicant's submission filed on 15 December 2006 has been entered.

### *Response to Arguments*

Applicant's arguments filed 15 December 2006 have been fully considered but they are not persuasive.

Applicant alleges:

> "...the Applicant notes that if one were to combine the Ishige and Melanson references, the combination cannot change the principle of operation of one of these references (see sect. 2143.03 of the MPEP). The Applicant submits that this would clearly happen if these references were combined."

Examiner disagrees. Applicant further states to substantiate this argument "The only information obtained from the user is the user's hearing characteristics, which are stored in the memory of the digital hearing aid when the hearing aid is **manufactured** (emphasis added)." However, Applicant has cited no portion of Ishige that teaches that

this is done during manufacturing of the hearing aid. In fact, Ishige teaches the opposite. The device is programmed after manufacture. This is shown in col. 6 lines 55 – 60 which states that the device is programmed during fitting, which occurs after manufacturing. Ishige further states in the embodiments relied upon for the rejections that the coefficients are programmable from an outside device or by using a removable ROM col. 8 lines 53 – 58.

Applicant goes on to state that "This is clearly a static operation, and does not allow the user to later provide a control signal to the hearing aid to dynamically control the performance of the hearing aid during operation." Again, this is contrary to what Ishige teaches. While Ishige does not actually teach a user operable control to change the characteristics, the reference is not limited to static operation. In col. 6 lines 55 – 60, Ishige teaches writing these fitting characteristics to the memory, or alternatively the characteristics may be stored in a removable ROM. Ishige clearly shows that the hearing aid device can have its characteristics changed via programming or the removable ROM further in col. 8 lines 53 – 58. While the device is static after it is set, this does not mean it is permanently static as Applicant alleges. The device can clearly be changed using the removable ROM or re-fitting **after manufacturing**.

Applicant then provides a characterization of the Melanson reference which the Examiner generally agrees with. However, Applicant then states "...if one were to combine these references, the principle operation of the Ishige reference would definitely change."

Applicant then provides a mis-characterization of what Applicant believes is the principle of operation of Ishige is, namely "Ishige teaches deriving coefficients for the transposed transversal filter based on a frequency analysis result of the input signal and the hearing characteristics of the user (**obtained by a static measurement and then programmed during manufacturing**), and using these coefficients to processes the input signal (bolded portions not taught). Ishige only envisions one processing strategy to process the input audio signal and does not even consider the possibility of altering the processin strategy based on user input."

While Ishige only provides one hearing compensation strategy after the device is fitted, it is by no means static as is stated above. Ishige includes all of the features necessary to allow for multiple processing strategies (a memory, a modifiable filter depending on the strategy). Melanson discloses these same features and adds an additional feature of allowing a user to change the compensation strategy depending on the environment. There is no doubt that it would be desirable to add this feature to the hearing aid of Ishige. It would aid a person with hearing loss in a variety of different environments. A simple modification of Ishige is all that is required (i.e. adding the various strategies into Ishge's memory and adding Melanson's user input).

Finally, it is submitted that Ishige's principle of operation is not what Applicant has suggested, but rather hearing aid device with an adjustable filter (Fig. 7, 22) that can allow a variety of hearing characteristics (via elements 25 – 27 of fig. 7) to be fitted using a memory or programming (Fig. 7, 31). MPEP 2143.01 requires that "suggested combination of references would require a substantial reconstruction and redesign of

the elements shown in [the primary reference] as well as a change in the basic principle under which the [primary reference] construction was designed to operate." It is submitted that there is no 'substantial reconstruction and redesign of the elements.' The most significant alterations are adding a user input and, perhaps not even necessary, adding a larger memory to allow for the multiple compensation characteristics. Furthermore, this modification would not change the basic principle under which Ishige operates.

Applicant further alleges that Ishige does not teach:

> "receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically adjusting compression characteristics for controlling the configuration of said input/output characteristic for loudness normalization."

Examiner disagrees. Applicant provides the details of compressor 202 in detail but fails to examine the references as a whole attempting to portray that compression is not changed via the user input. However, this is not the case.

It should be noted that the following was noted in the previous rejection. Melanson discloses allowing a user to dynamically select which dsp means to invoke in a given listening environment, one of the features that may be implemented is compression (with compression being italicized). In col. 19, it is clearly shown that these processing strategies allow for changing the compression strategies dynamically; col. 19 lines 10 – 15 (The digital signal processor presents the possibility of completely

changing the filter bank structures and compression strategies dynamically by loading different digital signal processing programs).

Applicant further alleges:

"the user adjustable digital loudness normalization control signal recited in claim 1 is used in a different manner and provides different results than the program selector switch taught by Melanson. The program selection switch of Melanson can only be used to select between a limited number of programs."

Applicant then further discusses the differences between the user adjustable control signals. However, in response to applicant's argument that the references fail to show certain features of applicant's invention, it is noted that the features upon which applicant relies (i.e., minimal measurements during an initial fitting, elimination f the time consuming and laborious task of measuring loudness data for the user) are not recited in the rejected claim(s). Although the claims are interpreted in light of the specification, limitations from the specification are not read into the claims. See *In re Van Geuns*, 988 F.2d 1181, 26 USPQ2d 1057 (Fed. Cir. 1993).

Furthermore, the benefits claimed do not differentiate the claimed limitations from the Melanson disclosure. Reading the claims as broadly as possible Melanson's program adjustable switch reads upon the user adjustable control signal.

Applicant further states with respect to claim 13:

"...the subject application recites receiving N user adjustable digital loudness normalization control signals from a user during operation for dynamically controlling a configurable composite/input/output characteristic for loudness normalization, each of the loudness control signals corresponding to one of the frequency domain input signals corresponding to one of a number of frequency domain input signals..." and "although Melanson teaches multi-band processing, the user is not allowed to separately control any parameters used in the different bands."

Examiner disagrees. First, the claim is not limited to separately controlling the parameters in different bands. Secondly, Melanson does allow for this. The user adjustable program selector varies all of the bands in the signals as is further evidences by the multi-band processing cited by Applicant.

***Claim Rejections - 35 USC § 103***

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

**Claim 1 – 40, 42, 43 and 45 - 57** are rejected under 35 U.S.C. 103(a) as being unpatentable over Ishige (U.S. Patent 5,892,836) in view of Melanson (U.S. Patent 6,104,822).

Regarding **Claims 1, 13 and 33**, Ishige discloses:

A method of generating an analog acoustic output signal from an acoustic input signal in accordance with a configurable input/output characteristic (abstract), said method comprising the steps of:

(a) converting the acoustic input signal into a digital acoustic input signal (i.e. and input circuit for receiving the analog audio signal and converting it into a digital signal; Fig. 7 element 12);

(b) transforming the digital acoustic input signal into one or more frequency domain input signals (Fig. 6 and col. 7 lines 37 – 57);

(c) detecting the magnitude of each of the one or more frequency domain input signals (i.e. a frequency analyzer; Fig. 7 element 21).

Ishige does not explicitly disclose:

(d) receiving a user adjustable digital loudness normalization control signal for dynamically adjusting compression characteristics for controlling the configuration of said input/output characteristic;

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal.

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value.

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal:

(h) converting the digital acoustic output signal into the analog acoustic output signal.

Melanson discloses a digital signal processor hearing aid with a program selector switch (Fig. 1a element 46) that is preferably manipulable by a user to allow the user to dynamically select which of the digital signal processing means to invoke in which listening environment. In dealing with these environments, each of the processing means may implement such functions as *compression*, noise compensation, feedback cancellation, etc; col. 8 lines 30 – 50.

Applying this environmental selection switch to Ishige would allow the user to conveniently alter the characteristic's of Ishige's hearing aid to further assist the user in various environments.

Modifying Ishige to include the selection feature taught by Melanson discloses:

(d) receiving a user adjustable digital loudness normalization control signal from a user during operation for dynamically adjusting compression characteristics for controlling the configuration of said input/output characteristic for loudness normalization (col. 8 lines 30 – 50; col. 19 lines 10 – 15 of Melanson; );

(e) for each of the one or more frequency domain input signals, determining a gain value in response to the user adjustable digital loudness normalization control signal and the magnitude of the frequency domain input signal (i.e. the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10; this portion of Ishige functions as the various DSPs in Fig. 1a of Melanson);

(f) providing one or more frequency domain output signals by multiplying each of the frequency domain input signals by the corresponding gain value (i.e. the digital audio signal is supplied to the hearing compensating circuit; Fig. 7 element 22);

(g) transforming the one or more frequency domain output signals into a digital acoustic output signal (i.e. the output of the hearing compensation circuit is applied to the output circuit; Fig. 7);

(h) converting the digital acoustic output signal into the analog acoustic output signal (Fig. 7 element 13).

It would have been obvious to one of ordinary skill at the time of the invention to apply the adjustable features of Melanson to the hearing aide of Ishige. Melanson discloses that previous hearing aids are typically capable of only providing one single strategy with adjustable parameters. This is similar to what Ishige discloses. Melanson further states that in the hearing aid of Fig. 1a, the hearing aid can implement more than one strategy and thus is better able to adapt and to provide optimal results in a variety of different listening environments.

Regarding **Claims 2, 14, 34 – 37 and 57**, in addition to the elements stated above regarding claims 1, 13, and 33, the combination further discloses:

wherein step (d) further comprises adjusting the configurable input/output characteristic for at least one frequency band corresponding to the one or more frequency domain input signals by one of:

increasing the level of said configurable input/output characteristic by a larger amount for lower level sounds compared to higher level sounds when a user adjusts the user adjustable digital loudness normalization control signal to increase the level of the analog acoustic output signal,

and decreasing the level of said configurable input./output characteristic by a smaller amount for lower level sounds compared to higher level sounds when the user adjusts the user adjustable digital loudness normalization control signal to decrease the level of analog acoustic output signal (i.e. in the combination, Ishige's hearing aid is adjustable as taught by Melanson. Melanson discloses that the invention is adjustable to different listening environments thus optimizing the amplification to fit the environment; cols 8 and 9. Thus, adjusting the hearing aid of the combination increases and decreases the level of the input/output characteristic dependent upon the setting of the user in various environments).

Regarding **Claims 3, 15 and 50**, in addition to the elements stated above regarding claims 1, 13 and 33, the combination further discloses:

performing steps (c), (e) and (f) by means of a programmable processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals and are programmed via the memory through the fitting device).

Regarding **Claims 4, 21, 39 and 53**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises calculating the corresponding gain value for the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor, wherein a parameter of the fitting formula is provided by the user adjustable digital loudness normalization control signal (i.e. the hearing compensating filter coefficient setting circuit defines the coefficients for the filters in the hearing compensating circuit on the data that is in the memory which is supplied by the user (Fig. 7 and the associated text in the disclosure) through the input of Melanson)

Regarding **Claims 5, 22, 38 and 51**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a look-up table stored in said programmable digital signal processor, wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 6, 23, 24 and 52**, in addition to the elements stated above regarding claims 5, 22 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claims 7, 25, 40 and 54**, in addition to the elements stated above regarding claims 3, 15, 34 and 50, the combination further discloses:

wherein step (e) comprises determining the corresponding gain value for each of the one or more frequency domain input signals by means of a fitting formula programmed into said programmable digital signal processor and a look-up table, wherein information in the look-up table is retrieved based on the user adjustable digital loudness normalization control signal (i.e. the coefficient table in Fig. 7; and at the time of changing the characteristics of the hearing compensating filter, the channel filter coefficient setting circuit refers to the coefficients stored in the coefficient table; col. 8 lines 45 – 55; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7; which is adjustable as taught by Melanson above).

Regarding **Claims 8, 16, 17, 26, 27 and 55**, in addition to the elements stated above regarding claims 7, 5, 25 and 50, the combination further discloses wherein said look-up table is stored in non-volatile memory in said programmable digital signal processor (i.e. Ishige discloses (*processor*) elements 21, 22, 24, 25, 26, 27 and 31 in Fig. 7 that process digital signals (*a digital signal processor*) and are programmed via the memory through the fitting device; and the coefficient table can be stored in ROM; col. 8 line 55).

Regarding **Claim 9**, in addition to the elements stated above regarding claim 1, the combination further discloses:

wherein step (b) comprises transforming the digital acoustic signal into at least two frequency domain input signals, each of said frequency domain input signals having a configurable channel input/output characteristic associated therewith, said configurable channel input/output characteristic together forming said configurable input/output characteristic, and wherein said at least two frequency domain input signals are provided with different channel input/output characteristics (i.e. Fig. 6 and col. 7 lines 37 – 57; and the hearing compensating filter coefficient circuit receives the analysis result and the hearing characteristics of the person and sets the filer coefficients; col. 7 lines 57 – 67 and col. 8 lines 1 – 10).

Regarding **Claims 10 – 12, 20, 28 – 30 and 45 - 49**, in addition to the elements stated above regarding claims 1, 13 and 34, the combination further discloses wherein

said configurable input/output characteristic is a curvilinear compression characteristic, an input compression characteristic, and an output compression characteristic. Ishige discloses matching the input audio signal with the narrowed dynamic range of the person fitted with the hearing aid using a filter; col. 7 lines 4 – 7. Thus, depending on the users hearing loss characteristics, the device may increase or decrease the high and low frequency components at different values. Further portions will lower amplitude values may be increased or decreased accordingly. As such, Ishige anticipates this element of the claimed invention. Additionally, the device is adjustable as taught by Melanson above.

Regarding **Claims 18, 19, 31 and 32**, in addition to the elements stated above regarding claims 9, 2, 13 and 33, the combination further discloses:

wherein each of said configurable channel input/output characteristics are (is) varied in response to said user adjustable digital loudness normalization control signal (i.e. the memory is programmed/fitted to the hearing characteristic of the user; and the hearing compensating circuit is configured to cause the input audio signal to match with the narrowed dynamic range of the person fitted with the hearing aid; col. 7 lines 4 – 7).

Regarding **Claim 56**, in addition to the elements stated above regarding claim 33, the combination further discloses:

(a) a microphone for receiving an input sound providing an analog input acoustic signal (Fig. 7 element 11)

(b) an A/D converter coupled to said sound energy signal and for reception device for receiving said analog input acoustic signal or an image of said analog input acoustic signal and coupled to said analysis filter for providing said digital acoustic input signal (Fig. 1 element 12)

(c) a D/A converter coupled to said synthesis filter for receiving said digital output acoustic signal and for providing an analog output acoustic signal (Fig. 1 element 13);

(d) a speaker coupled to said D/A converter for receiving said analog output acoustic signal and providing an output sound energy signal (Fig. 1 element 14).

Regarding **Claims 42 and 43**, claims 42 and 43 claim various methods of which to adjust the control signal, a variable resistor and a two-way switch which are not explicitly disclosed by the combination. However, Melanson discloses a user manipulable switch but does not provide details on its implementation.

Examiner takes official notice that various different switches exist including variable resistors (i.e. potentiometers as shown by Martin US 6,104,822 previously) and mechanical switches.

As Melanson is silent it is clear that the implementation of the switch is not a key feature. Many different switches exist and their implementations in the art of hearing aides (such as the one of the combination) do not produce any new or unexpected results. In other words, using a mechanical two-way or a potentiometer would not patentably distinguish the claimed invention from the prior art because it does not produce any new or unexpected result.

Thus implementing a programming means such as the method disclosed by Martin or a two way switch would have been obvious to one of ordinary skill in the art. One would have been motivated to do so to in order to effectively enter an input in to the combination's hearing aid.

### *Conclusion*

All claims are drawn to the same invention claimed in the application prior to the entry of the submission under 37 CFR 1.114 and could have been finally rejected on the grounds and art of record in the next Office action if they had been entered in the application prior to entry under 37 CFR 1.114. Accordingly, THIS ACTION IS MADE FINAL even though it is a first action after the filing of a request for continued examination and the submission under 37 CFR 1.114. See MPEP § 706.07(b). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

It should be noted that the amendment to claim 1 of adding "adjusting compression characteristics for" was met in the previous final rejection on the bottom of page 5 and therefore could have finally been rejected.

**THIS ACTION IS MADE FINAL.** Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the

shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the mailing date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Andrew C. Flanders whose telephone number is (571) 272-7516. The examiner can normally be reached on M-F 8:30 - 5:00.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Sinh Tran can be reached on (571) 272-7546. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

acf

SINH TRAN
SUPERVISORY PATENT EXAMINER